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Recommendation

ITU-T H.705.2 (09/2023)

SERIES H: Audiovisual and multimedia systems

IPTV multimedia services and applications for IPTV – General aspects

Requirements for live streaming systems based on the QUIC transport protocol

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Recommendation ITU-T H.705.2

Requirements for live streaming systems based on the QUIC transport protocol

Summary

Recommendation ITU-T H.705.2 specifies the requirements of a live streaming system to utilize quick user datagram protocol (UDP) Internet connections (QUIC) transport protocol to improve its delivery performance and security. It also describes the procedures and framework for QUIC-based live streaming system to provide unicast or multicast service encapsulating in QUIC protocol.

With this Recommendation, a live streaming service provider can gain an understanding of how to utilize QUIC protocol to provide unicast or multicast live streaming media service. With QUIC transport protocol, the services will have lower connection establishment and delivery delay, enhanced delivery performance, and security insurance.

History [*]

Edition	Recommendation	Approval	Study Group	Unique ID
1.0	ITU-T H.705.2	2023-09-13	16	11.1002/1000/15634

Keywords

CDN, HTTP, live streaming, multicast, QUIC, RTP, RTSP.

[*] To access the Recommendation, type the URL https://handle.itu.int/ in the address field of your web browser, followed by the Recommendation's unique ID.

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In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

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1 Scope

This Recommendation focuses on solving the problems that emerge when deploying quick user datagram protocol (UDP) Internet connections (QUIC) on a live streaming system, as well as systematically analysing the QUIC streaming system requirements and use case scenarios.

The Recommendation covers the study of:

– Use case studies and QUIC-based live streaming scenarios analysis;
– Requirements on QUIC-based live streaming platforms;
– Requirements on QUIC-based live streaming network, including multicast requirements and device requirements;
– Requirements on QUIC-based live streaming system architecture evolution including real-time transport protocol (RTP) mapping and hybrid request routing;
– Security mechanisms requirements.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published. The reference to a document within this Recommendation does not give it, as a stand-alone document, the status of a Recommendation.

[ITU-T Y.1901] Recommendation ITU-T Y.1901 (2009), *Requirements for the support of IPTV services*.

[ITU-T Y.1910] Recommendation ITU-T Y.1910 (2008), *IPTV functional architecture*.

[IETF RFC 9000] IETF RFC 9000 (2021), *QUIC: A UDP-Based Multiplexed and Secure Transport*.

[IETF RFC 9001] IETF RFC 9001 (2021), *Using TLS to Secure QUIC*.

3 Definitions

3.1 Terms defined elsewhere

This Recommendation uses the following terms defined elsewhere:

3.1.1 application level gateway (ALG) [b-ETSI TS 123 228]: An application specific functional entity that allows communication between disparate address realms or IP versions, e.g., an IPv6 node to communicate with an IPv4 node and vice versa, when certain applications carry network addresses in the payloads like SIP/SDP. NA(P)T-PT or NA(P)T is application unaware whereas ALGs are application specific translation entities that allow a host running an application to communicate transparently with another host running the same application but in a different IP version or IP address realm.

3.1.2 content delivery network (CDN) [b-ITU-T Y.2084]: A content delivery network (CDN) is a system of distributed servers that deliver content (e.g., web pages, files, videos and audios) to users based on pre-defined criteria such as the geographic locations of users, the status of the content delivery server and the IP network connection.

3.1.3 Internet protocol television (IPTV) [ITU-T Y.1901]: Multimedia services such as television/video/audio/text/graphics/data delivered over IP-based networks managed to support the required level of QoS/QoE, security, interactivity and reliability.

3.1.4 terminal device (TD) [ITU-T Y.1901]: An end-user device that typically presents and/or processes the content such as a personal computer, a computer peripheral, a mobile device, a TV set, a monitor, a VoIP terminal, or an audiovisual media player.

3.2 Terms defined in this Recommendation

This Recommendation defines the following terms:

3.2.1 hybrid QUIC-based live streaming system: A QUIC-based live streaming system that only uses the QUIC protocol on some but not all of the uploading, delivery, and streaming service phases to transport live streaming content.

3.2.2 over-the-top live streaming: A live streaming service whose delivery process relies on the Internet rather than the traditional Internet protocol television (IPTV) dedicated network.

4 Abbreviations and acronyms

This Recommendation uses the following abbreviations and acronyms:

CDN	Content Delivery Network
DASH	Dynamic Adaptive Streaming over HTTP
GSLB	Global Server Load Balancer
HLS	HTTP Live Streaming
HTTP	Hypertext Transfer Protocol
IPTV	Internet Protocol Television
QUIC	Quick UDP Internet Connections
RTP	Real-time Transport Protocol
RTMP	Real-time Messaging Protocol
RTSP	Real-time Streaming Protocol
STB	Set-Top Box
TCP	Transmission Control Protocol
TLS	Transport Layer Security
UDP	User Datagram Protocol

5 Conventions

In this Recommendation:

– The keywords "is required to" indicate a requirement which must be strictly followed and from which no deviation is permitted if conformance to this document is to be claimed.

– The keywords "is recommended" indicate a requirement which is recommended but which is not absolutely required. Thus, this requirement need not be present to claim conformance.

6 Introduction

The existing live streaming systems usually use real-time streaming protocol (RTSP) on Internet protocol television (IPTV) live streaming media service or hypertext transfer protocol (HTTP) on over-the-top live streaming (OTT live streaming) media service. IPTV live streaming system can use real-time transport protocol (RTP) over UDP on multicast services, and RTP over user datagram protocol (UDP) / transmission control protocol (TCP) on unicast service. However, OTT live streaming systems are running on the Internet, which is not a multicast dedicated network, and therefore cannot use the multicast protocol for its live content delivery. It usually uses content delivery network (CDN) to relay its unicast streaming requests for providing live streaming service, by using HTTP over TCP as its transport protocol. Those two types of existing streaming systems that adopt the TCP/UDP transport protocol do not meet the low latency and high quality requirements of live streaming service very well, albeit with multiple supplemental techniques provided. Because TCP is implemented in operating system kernels and client box firmware, making significant changes to TCP is next to impossible.

Quick UDP Internet connections (QUIC) [IETF RFC 9000] is a new transport layer protocol that reduces latency compared to that of TCP. It is very similar to TCP+TLS+HTTP/2 implemented on UDP. Since QUIC is built on top of UDP, it is very easy to be implemented on an IPTV set-top box (STB) or a PC. In the current ITU-T Recommendations, there is a lack of study of the use cases and scenarios for using QUIC to improve the quality of delivering live content on an unreliable network with low latency requirements.

6.1 Introduction to the QUIC protocol

IETF QUIC covers different aspects of the 'core' QUIC protocol, defines mapping of HTTP semantics to it, and covers selected other topics such as transport, recovery, transport layer security (TLS), HTTP, QPACK, and load balancers.

The key features of QUIC include:

– Dramatically reduced connection establishment time.

– Improved congestion control.

– Multiplexing without head of line blocking.

– Connection migration based on the situation.

– Better encryption performance.

Due to the above advantages of QUIC protocol, a growing number of applications have started to use it. However, there are multiple problems that need to be addressed before massive deploying QUIC on live streaming, including:

1) Switches, gateways, and firewalls may block UDP ports used by QUIC.

2) Due to the QoS restriction, Internet service providers may consider exceeding UDP packets as DDoS attacks therefore dropping them.

3) The standards for deploying QUIC on multicast are not well defined.

4) Deploying QUIC may require several modifications to the current IPTV platform architecture, including service discovery, security mechanisms, CDN transmission and distribution, STB software, and so on.

6.2 QUIC-based live streaming system architecture

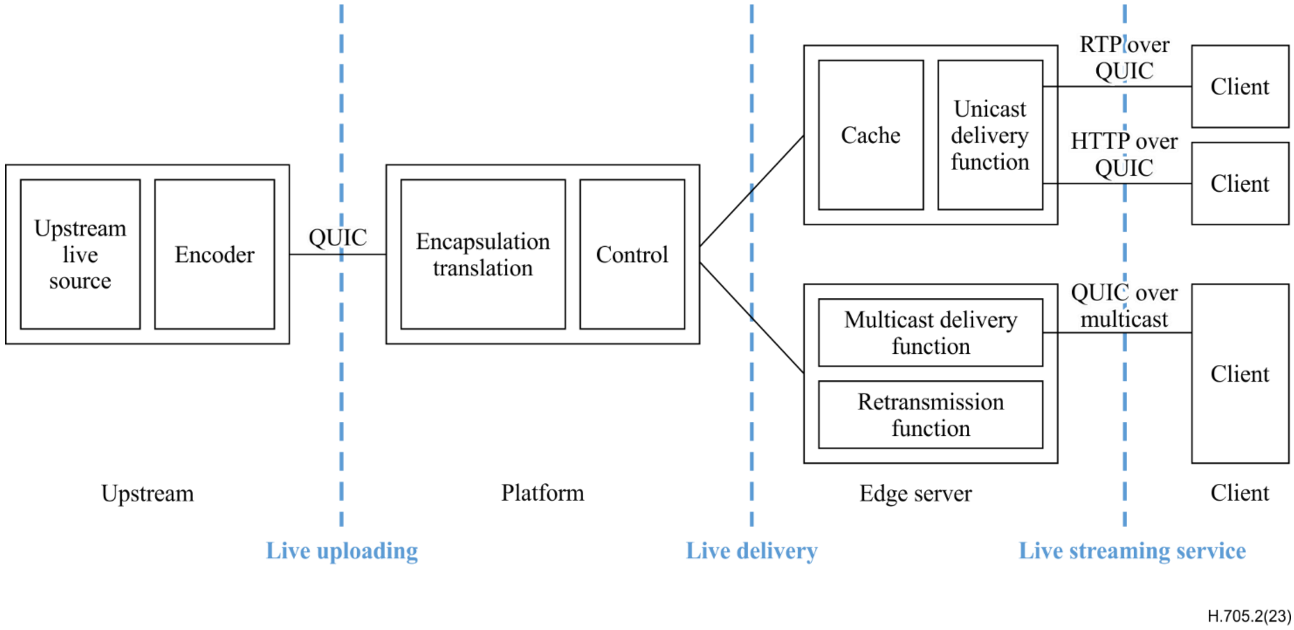

Figure 6-1 – QUIC-based live streaming architecture

QUIC-based live streaming can reuse the current application layer protocol, such as RTP/RTSP for IPTV or HTTP for OTT live streaming, while replacing the transportation layer protocol with QUIC.

– QUIC-based upstream source usually uploads media contents with HTTP 3.0 using OTT live streaming unicast.
– QUIC-based live streaming delivers content through HTTP 3.0. Since QUIC-based multicast technology is immature, most live streaming requests can be served through unicast.
– QUIC-based live streaming can utilize RTP/RTSP/HTTP 3.0 as its application protocol which uses QUIC as an underlying transport protocol.

7 Requirements on QUIC-based live streaming system

Aside from the requirements described in [ITU-T Y.1901], the QUIC-based IPTV service platform should also satisfy the following requirements when utilizing QUIC.

7.1 General requirements for QUIC-based live streaming service

GR-01: The QUIC-based live streaming service is required to support QUIC-based streaming service discovery.

GR-02: The QUIC-based live streaming service is required to support QUIC session control.

GR-03: The QUIC-based live streaming service is required to support QUIC security mechanism.

GR-04: The QUIC-based live streaming service is required to support QUIC packet forwarding.

GR-05: The QUIC-based live streaming service is required to support either QUIC unicast or QUIC based multicast.

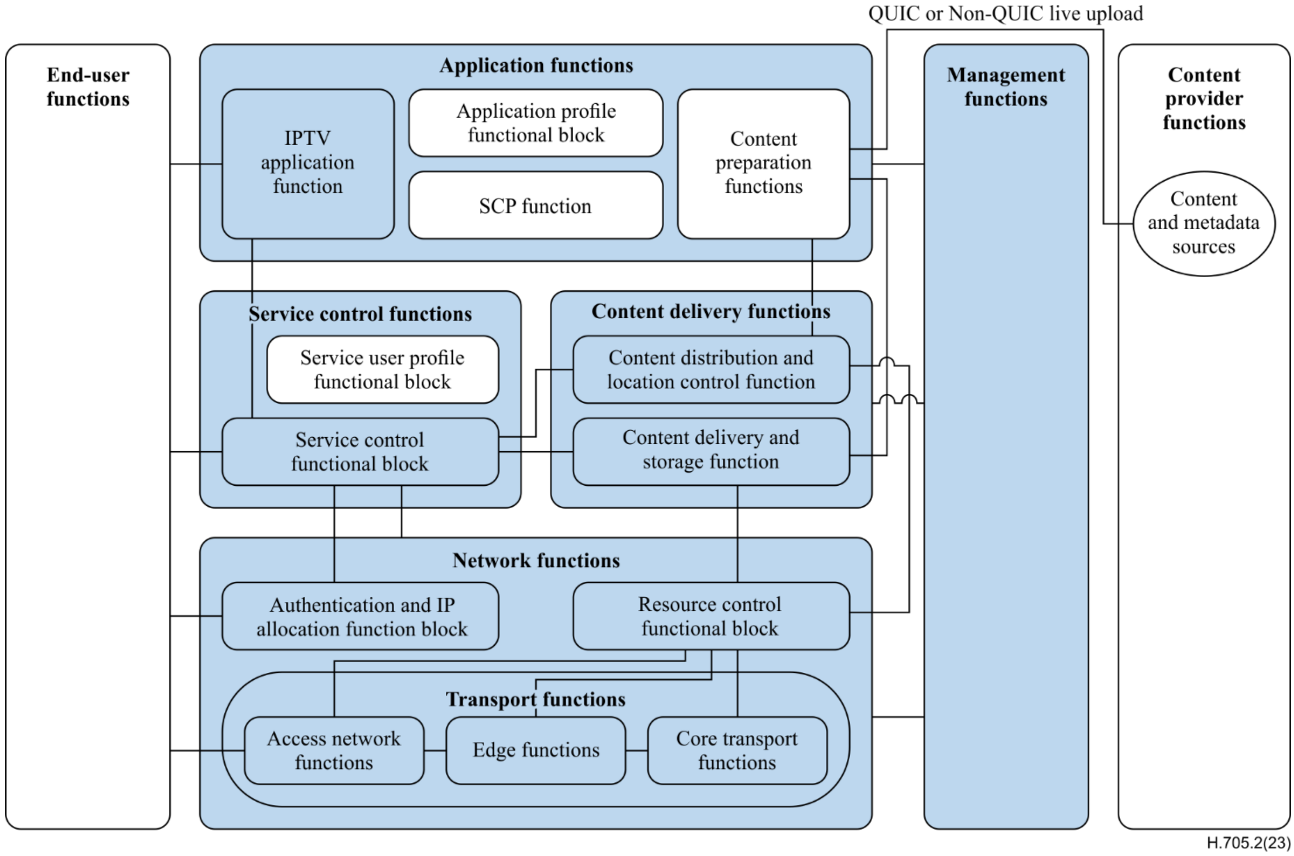

Figure 7-1 – Functions of QUIC-based IPTV live streaming system

7.2 Requirements for application functions

RAF-1: The application functions are required to support QUIC-extended service control function API.

7.3 Requirements for service control functions

RSCF 1: The service control functions are required to support QUIC-extended content delivery functions API.

7.4 Requirements for content delivery functions

RCDF 1: The content delivery functions are required to support QUIC transport protocol based on TLS/1.3. This Recommendation is based on TLS/1.3 without the guarantee of compatibility with future TLS versions. Future TLS version support is recommended but not required.

RCDF 2: The content delivery functions are required to support QUIC-based client requests.

RCDF 3: The content delivery functions are recommended to distribute content using QUIC multicast.

RCDF 4: The content delivery functions are recommended to support QUIC encryption with external keys provided by service control functions.

RCDF 5: The content delivery functions are recommended to support QUIC unicast and QUIC multicast services.

RCDF 6: The content delivery functions are recommended to retransmit multicast content using QUIC unicast connection.

RCDF 7: The content delivery functions are recommended to support QUIC multicast session control.

RCDF 8: The content delivery functions are recommended to support QUIC multicast session secret key generation and distribution to the client.

RCDF 9: The content delivery functions are recommended to support the encapsulation of existing protocols using QUIC, for example, RTP over QUIC.

RCDF 10: The content delivery functions are recommended to support QUIC-based request routing services.

RCDF 11: The content delivery functions are recommended to support user IP migration based on QUIC connections.

7.5 Requirements for network transport functions

RNF 1: The transport functions are required to recognize QUIC-encapsulated UDP packets.

RNF 2: The transport functions are required to forward QUIC-encapsulated UDP traffic based on configurations.

RNF 3: The transport functions are required to allow QUIC-encapsulated UDP packets to pass the application level gateway (ALG) if the ALG is presented.

7.6 Requirements for management functions

RMF 1: The management functions are required to configure QUIC requirements and recommendations in clause 8.

8 Adaptations for QUIC based IPTV functions

To satisfy the general functional requirements described in clause 7.1, some IPTV functions should be extended to have augmented functionalities concerning QUIC.

The client can detect HTTP-based QUIC streaming services such as HTTP live streaming (HLS), dynamic adaptive streaming over HTTP (DASH) with HTTP alternative services. If QUIC service is available, the client can send the QUIC-based service request to the IPTV system.

8.1 Adaptations for application functions

According to [ITU-T Y.1910], the application functions enable the end-user functions to select and purchase or rent a content item.

Application functions do not need to change to accommodate QUIC delivery. The data uploading from the content provider functions and content delivery to the end-user functions via QUIC are handled by the network functions, and the application functions will not be concerned about QUIC related issues. It only needs to be extended with an API for QUIC protocol selection.

8.2 Adaptations for service control functions

According to [ITU-T Y.1910], the IPTV service control functional block provides the functions to handle service initiation, modification and termination requests, perform service access control, and establish and maintain the network and system resources required to support the IPTV services requested by the IPTV terminal functions.

The service control functions authenticate and redirect the user request to content delivery functions, and therefore they should support the QUIC-based service API.

8.3 Adaptations for content delivery functions

According to [ITU-T Y.1910], the content delivery functions (CDF) perform cache and storage functionalities and deliver the content according to the request from the end-user functions.

The content delivery functions are the cardinal functions to implement and apply the QUIC protocol as the underlying transportation protocol. They handle QUIC session initialization, manage QUIC connections, and distribute QUIC keys in multicast service. Also, the user IP migration can be implemented in content delivery functions to fully facilitate QUIC features.

It is possible for the content delivery functions to repair multicast content with unicast transmission with either or both of the methods being QUIC-based.

8.4 Adaptations for network functions

The network functions are the basic infrastructure of the QUIC protocol. They should allow the QUIC-encapsulated UDP traffic to be delivered.

8.5 Adaptations for management functions

The management functions are required to be able to configure the IPTV service platform to support QUIC functionalities as illustrated above.

9 QUIC-based live streaming framework

9.1 Hybrid QUIC-based live streaming system

Although QUIC-based live streaming system is more efficient, secure, and reliable, and may be a better solution for future live streaming systems, the majority of extant Internet equipment does not support QUIC and thus requires software or hardware updates regarding service platform, CDN, network system, and terminal devices [b-ITU-T H.720]. The evolution towards the QUIC protocol is a time-consuming progress, in which most live streaming systems will be hybrid live streaming systems for an extended period.

In a hybrid QUIC-based live streaming system, the uploading, delivery, and streaming service phases can independently use the QUIC protocol to transport their content. It is required to deploy the QUIC protocol translation function on the boundary between QUIC-enabled transportation components and QUIC-disabled transportation components. The clauses below describe exemplary hybrid QUIC-based live streaming systems.

9.1.1 Hybrid QUIC/IPTV-based live streaming architecture

Since QUIC multicast technology is immature, the IPTV live streaming system should use the traditional multicast method during the delivery phase and use QUIC-based transmission during live streaming service phase.

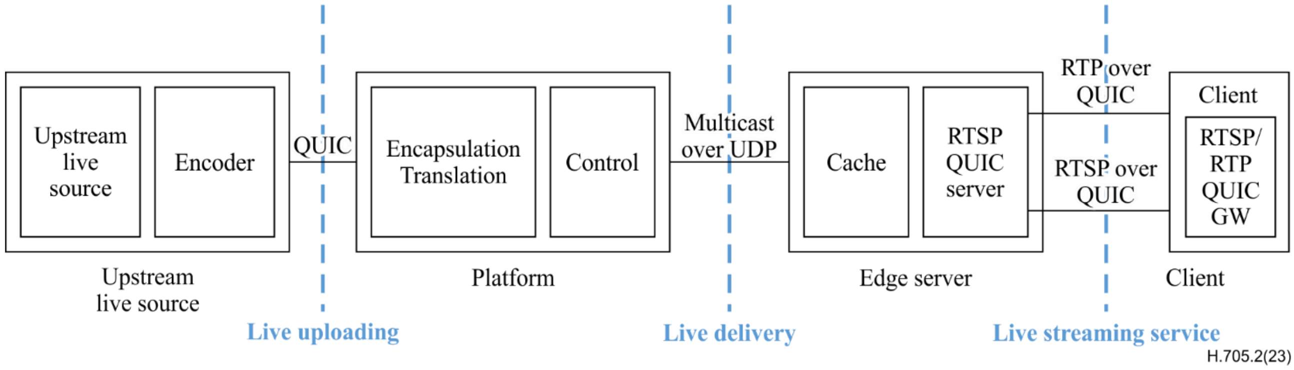

Figure 9-1 – Hybrid IPTV multicast system using QUIC

In the hybrid IPTV live streaming transmission architecture exhibited in Figure 9-1, the edge server and the client are extended with the QUIC protocol translation module. The client initiates an RTSP request through the QUIC protocol translation module to the edge RTSP live streaming server, which receives and translates the QUIC encapsulated message, generates the corresponding RTSP response encapsulated within the QUIC protocol, and sends the response via QUIC connection to the client to be processed. Then the edge server starts transmitting the requested media by translating RTP-based multicast media contents that are already received from the central node into the QUIC-based unicast content and sends them to the client. The client uses the QUIC protocol translation module to translate the QUIC-based packets into RTP-based media streams which can be recognized and processed by most media players.

Clients that do not support the QUIC protocol can utilize the extant multicast protocol with the retransmission mechanism based on QUIC in order to improve the QoS of the live streaming service.

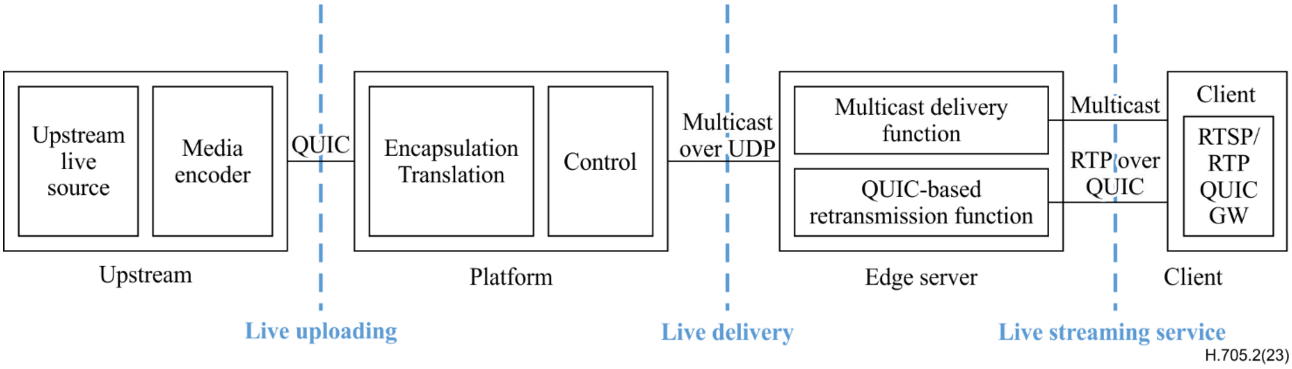

Figure 9-2 – Hybrid IPTV live streaming interfaces

To re-transmit data in multicast, the live streaming system should have a retransmission function module that can cache the multicast content and its metadata in the memory from the upstream multicast source server. When the client detects a transmission error, it establishes a QUIC-based connection with the live stream system retransmission function, which may not reside in the same physical nodes as the multicast delivery function does, and then sends a retransmission request. The retransmission function receives the retransmission request, locates the referred content using metadata, and then sends the requested content to the client through the established QUIC-based connection in a unicast manner. To detect packet missing or errors, the client QUIC protocol translation module should maintain a cache to arrange and sort the real-time transport protocol (RTP) contents.

9.1.2 Hybrid QUIC/OTT live streaming architecture

Without standardized QUIC multicast technology, QUIC-based OTT live streaming service still uses unicast. By introducing extant multicast networks and technology from IPTV into QUIC-based OTT live streaming systems in order to construct a hybrid live streaming system, the network bandwidth consumption in a streaming can be greatly reduced.

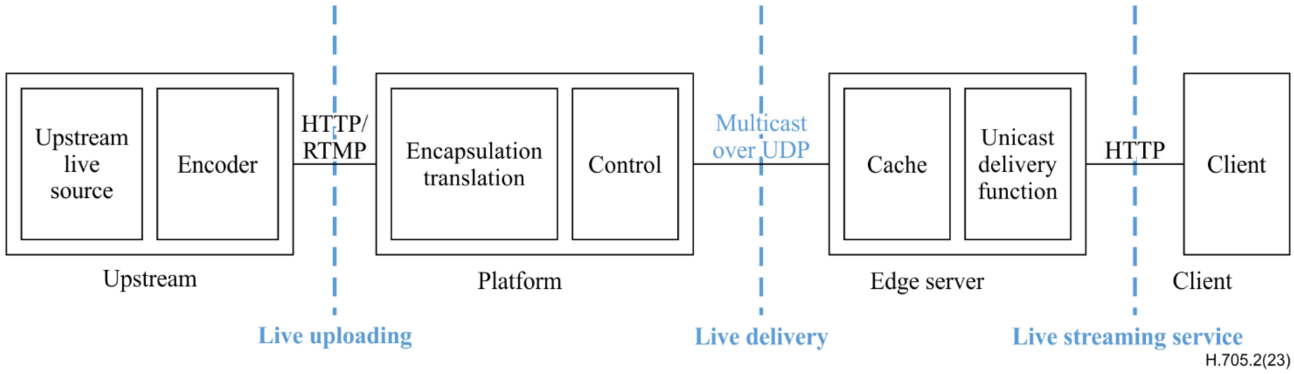

Figure 9-3 – OTT live streaming interfaces

In a QUIC-based OTT live streaming system, the platform can use encapsulation translation and transcoding functions to transform upstream media contents into multicast media contents that have a normalized encapsulation and coding scheme, and deliver them to the edge nodes through multicast. The edge nodes cache the content and provide live streaming service through QUIC-based unicast.

9.2 QUIC-based multicast service framework

Although there is no QUIC multicast profile being standardized or universally applied, there are proposals on the road towards standardization reflecting conjectures about future development. Their technical details may vary in future modifications, but the general design principles shall persist.

9.2.1 QUIC-based multicast streaming service

QUIC-based multicast streams can be delivered like standard multicast streams since they are all based on the user datagram protocol (UDP). However, the client cannot directly obtain media content encapsulated in the QUIC multicast stream by receiving the multicast stream, because it does not have the required key material to decrypt the received QUIC stream. The key material includes TLS state (encryption state, AEAD function information, etc.) and QUIC state (connection information, packet number, header protection key, etc.), which can be referenced in [IETF RFC 9001] and its referenced standards. To allow such decryption, there are two possible methods for QUIC-based multicast stream recipients to access the key material. A practice to define the QUIC multicast profile can be referenced in the draft "Multicast Extension for QUIC" [b-draft-jholland-quic-multicast].

1. The QUIC multicast service provider can agree with its clients on the key material of the multicast QUIC stream. The key material can either be explicitly defined or be derived through a set of rules.
2. The QUIC multicast service provider can send the key material to the client directly or indirectly. In a simplified scenario, the client can establish a QUIC connection with the QUIC multicast server and request the key material of the multicast channel. Upon receiving the key material, the client can therefore receive and decrypt the multicast QUIC stream.

The first method is simple but invalidates the meaning of using encryption and QUIC. The second option is more sophisticated but preferable if using QUIC multicast.

Using the second method, a potential service process can be described as follows. The client sends its content request with a content ID to the global server load balancer (GSLB), and the GSLB responds with multicast channel information for that content along with the key material. The client will join the multicast group using the multicast channel information and decrypt the QUIC stream using key material from GSLB. If the key material is changed periodically to ensure service management and safety, when it is updated, the GSLB will send a notification and new key material to the client if it is still active.

9.2.2 QUIC-based multicast service for unicast client

In some IPTV systems, clients do not have access to a multicast network due to ISP restrictions, but their closest gateway can access multicast content and relay that content to the client via unicast. In a QUIC-based IPTV system, the encryption process of QUIC must be taken into account.

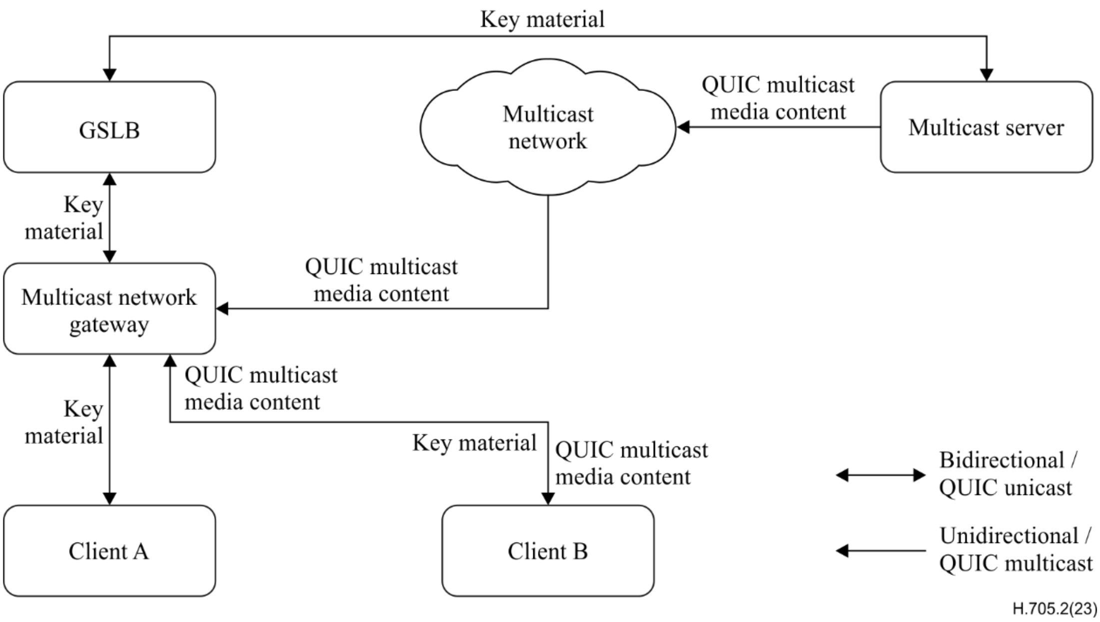

Figure 9-4 – Procedure of QUIC-based multicast streaming for unicast client

In this scenario, a client will first send the media content request with the content ID to the GSLB. The GSLB parses the request and responds with a multicast service label. The client will parse the multicast service label to obtain the QUIC-based multicast service information, including the QUIC multicast service name and use the QUIC multicast service information to request the multicast content from the gateway of the multicast network. The multicast service name will be used to locate the gateway through multicast DNS service discovery. Upon receiving the client's request, if the gateway has not received such a multicast label before, it will start the QUIC multicast service registration process. First, it retrieves QUIC multicast channel information, including key material, from the GSLB using the multicast label, and sends the key material to each requester. Then the gateway will join the multicast group of QUIC. When the gateway receives a QUIC multicast stream, and relays one copy of the stream to each requester through unicast. Each requester will use the key material to decrypt the QUIC stream and obtain the media content. When the key material is updated, the GSLB will send a notification and the new key material to the gateway, which in turn relays it to any of its active requesters for further usage.

Appendix I

Live streaming use cases and scenarios

(This appendix does not form an integral part of this Recommendation.)

Live streaming services can be classified from various perspectives. They can be divided into IPTV live streaming, OTT live streaming, and mobile device live streaming in terms of the types of terminal devices; live media streaming and whole file streaming in terms of the media formats; and unicast streaming, multicast streaming, and broadcast streaming in terms of the service network types. This Recommendation focuses on the live streaming system requirements based on QUIC, whose prominent feature is the low delay in transportation. Therefore, the following clauses will categorize the live streaming service into high-delay/latency live streaming, low-delay/latency live streaming, and ultra-low-delay/latency live streaming.

I.1 High latency live streaming (End-to-end delay > 5 seconds)

High latency live streaming usually applies to non-interactive live streaming scenarios such as non-interactive OTT live streaming and digital audio broadcast channels.

1) Typical end-to-end content delivery scenario

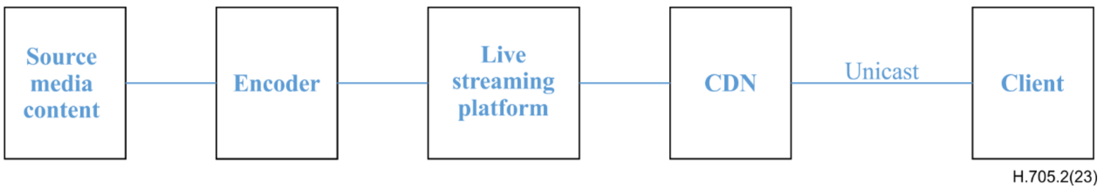

Figure I.1 – Typical end-to-end content delivery system for high latency live streaming

In a typical non-interactive high-latency live streaming scenario, such as UGC live streaming, the media host records and encodes the media contents and uploads them to the live stream platform, which in turn performs transcoding and multiplexing on the content and injects them into the content delivery network (CDN). The client can request the media contents from the CDN.

Due to the high latency during the content delivery, the high-latency live streaming service is only used in non-interactive live streaming scenarios and is unable to support the real-time interaction between the host and audiences, which is necessary for a live show.

2) Media format

Usually in high latency live streaming services, the payloads of media content are encapsulated in variable bit-rate container/file formats. The delivery of high latency live streaming content can be based on HTTP for session control and media data transmission, such as HLS and DASH.

3) Transportation protocol

The conventional high latency live streaming service usually delivers their contents through CDN via TCP protocol. In recent years, multicast OTT live streaming solutions have been gradually applied, which deliver HLS/DASH based media content through a multicast-dedicated network based on the UDP.

I.2 Low latency live streaming (End-to-end delay 1 ~ 5 seconds)

As the "live streaming + X" business models continue to be developed in various businesses, plenty of interactive live streaming applications emerge, such as e-commercial live advertisement, online education, live sports, and live entertainment, which put serious demands on reducing the streaming transmission delay.

Typical scenarios may include:

- E-commercial: in the "instant buy" activities, the high delay will reduce the initiatives to interact, thus lowering the sales.
- Online education: the interactive experience in remote teaching is highly dependent on the delay.
- Live sports: delay affects the psychological resonance of the audiences.
- Live show: delay affects the efficacy of interactions and feedback between the compère and the audiences.

Due to the demand for interactive capability, live streaming services require a better network performance with respect to low latency.

1) Typical end-to-end content delivery scenario

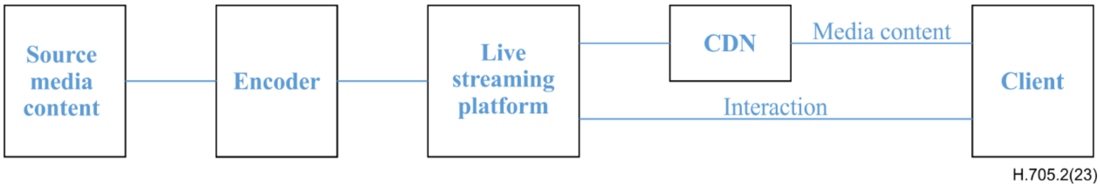

Figure I.2 – Typical end-to-end content delivery system for low latency live streaming

In a typical low latency live streaming scenario, the media source content, being locally encoded by the media content producer is usually uploaded via a streaming protocol such as a real-time messaging protocol (RTMP) [b-IETF RFC 7425] and WebRTC [b-IETF RFC 8835] to the live streaming platform, which will subsequently transcode and encapsulate the media content, and inject them onto the CDN. The client can request the media content using live media streaming protocol (RTMP and WebRTC) from the CDN.

In this scenario, the clients can interact with the live streaming platform and thus the host within a negligible delay in the sense of human communication, but they do not need to interact with the CDN.

2) Media format

In low latency live streaming, media content is usually encoded into a variable bit-rate version and encapsulated using stream media formats supported by live streaming transport protocols such as RTMP/TCP and RTP/UDP.

3) Network and protocol

Low latency live streaming usually distributes media content through the CDN, using TCP or UDP unicast network protocol.

I.3 Ultra-low latency live streaming (End-to-end delay < 1s)

The ultra-low latency live streaming is usually required for strong interactive live streaming scenarios, including field-of-view (FOV) enhanced virtual reality streaming, and cloud game streaming.

1) Typical end-to-end content delivery scenario

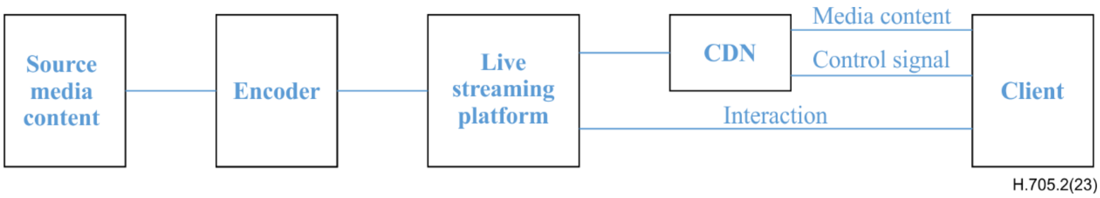

Figure I.3 – Typical end-to-end content delivery system for ultra-low latency live streaming

Taking viewport-dependent virtual reality (VR) live streaming as an example of the ultra-low latency live streaming scenario. In the viewport-dependent VR live streaming service, the VR content source will be stored in the CDN in the form of tiles, and the client will request the corresponding tiles via a control signal based on the viewport orientation to avoid downloading a full amount of VR media content. The quality of experience of viewport-dependent VR service highly relies on network latency.

In this scenario, the clients can interact with the live streaming platform using the control signal and the CDN within a negligible delay in the sense of human perception.

2) Media format

Ultra-low latency live streaming usually uses stream media formats that can have media content's random-access interval be less than a second, such as DASH and other stream media formats supported by RTP.

3) Network and protocol

Ultra-low latency live streaming usually distributes media content through the CDN, using UDP unicast streaming protocol which allows a stateful connection between the client and the CDN.

Appendix II

Conventional live streaming architecture

(This appendix does not form an integral part of this Recommendation.)

II.1 IPTV-based live streaming architecture

Figure II.1 shows the content delivery functions in the IPTV architecture and the corresponding reference points (refer to [ITU-T Y.1910] for further details).

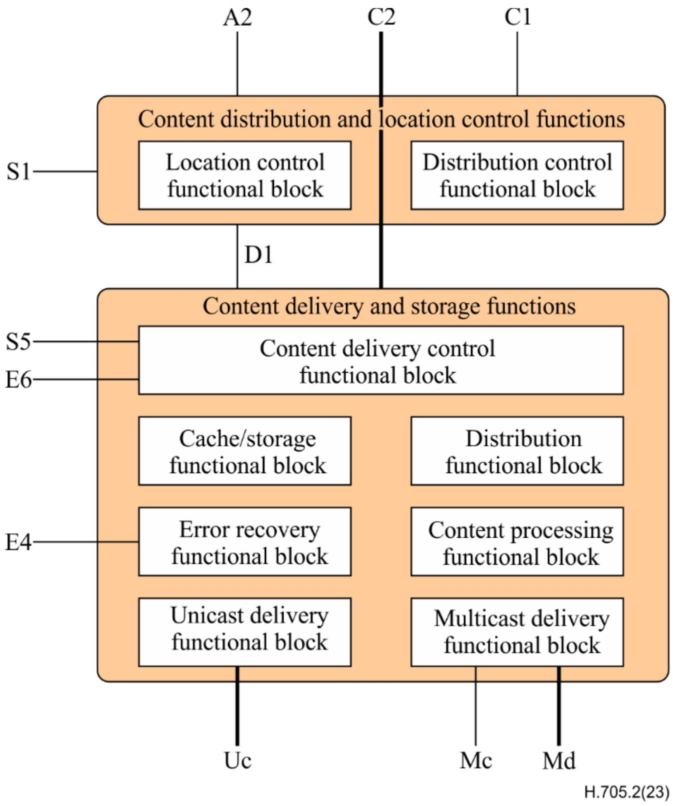

Figure II.1 – Content delivery functions in [ITU-T Y.1910]

In [ITU-T Y.1910], the content delivery functions consist of two groups of functions:

- Content distribution and location control functions (CD & LCF);
- Content delivery and storage functions (CD & SF).

Figure 7-1 applies to IPTV live streaming and video-on-demand scenarios. IPTV video on-demand service only use unicast transmission while live streaming services can use both unicast and multicast transmission.

According to the architecture exhibited in Appendix I, IPTV live streaming architecture can be depicted as the following:

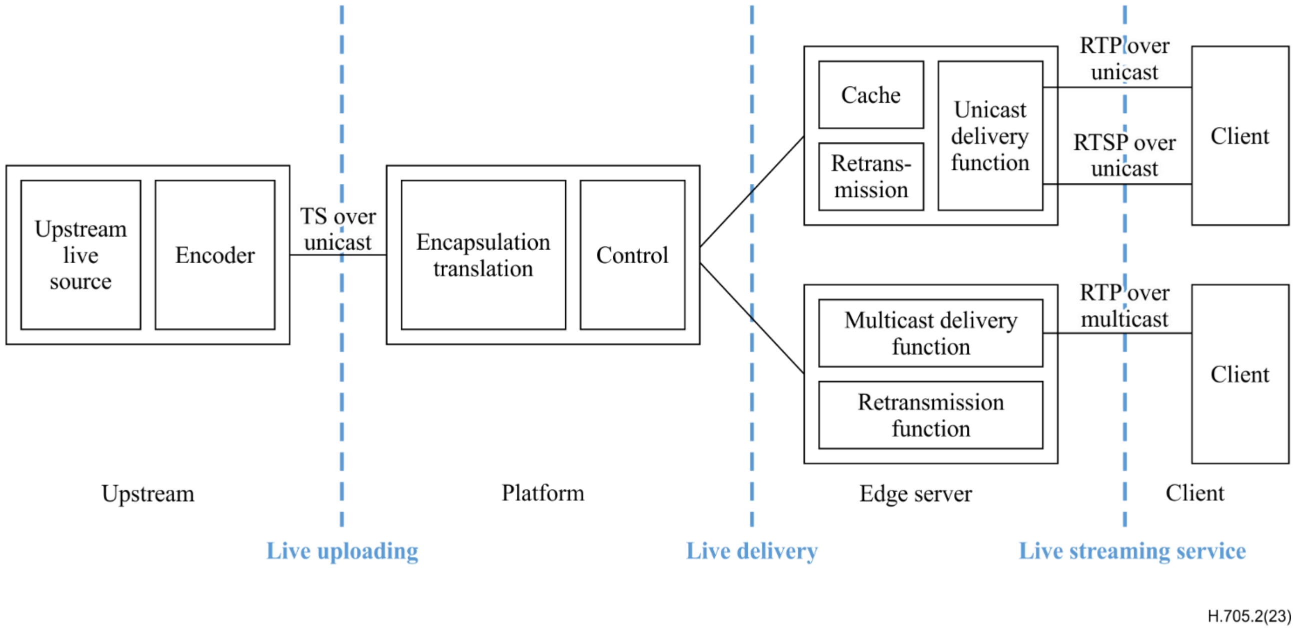

Figure II.2 – IPTV-based live streaming architecture

As Figure II.2 shows, IPTV live streaming consists of three phases classified as live streaming upload, live streaming distribution, and live streaming service.

– IPTV live streaming upload usually encapsulates the payloads through MPEG2-TS ([b-ISO/IEC 13818-1]) and transports them using unicast protocol through QoS guaranteed private network.

– IPTV live streaming distribution can deliver content through the CDN or Internet multicast via unicast or multicast separately.

– IPTV live streaming service usually adopts stream media method, using real-time streaming protocol (RTSP) as the control protocol and RTP as the data transmission protocol. Because RTP is built on top of UDP, the IPTV system usually has a retransmission function to improve QoS. IPTV live streaming can adopt both unicast and multicast methods.

II.2 OTT live streaming architecture

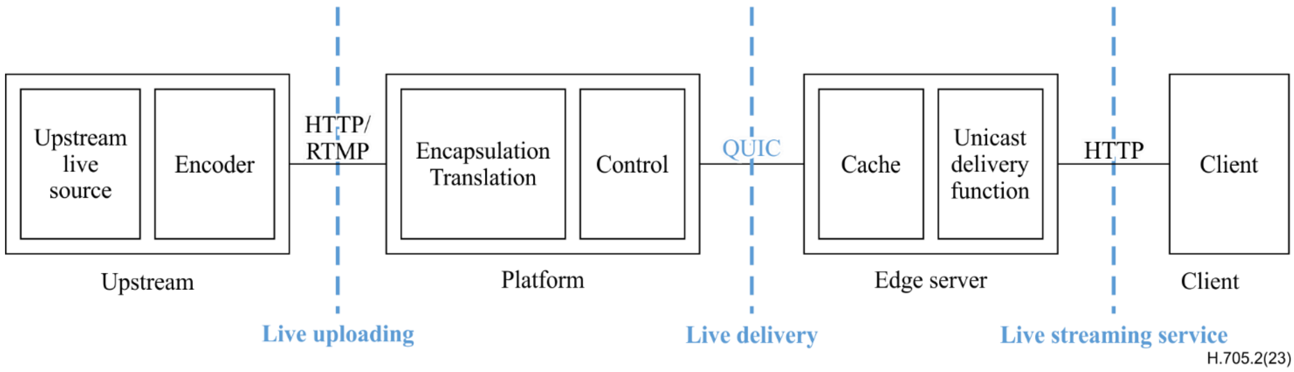

Figure II.3 – OTT live streaming architecture

Classical OTT live streaming consists of three reference points: live streaming upload, live streaming distribution, and live streaming service.

– OTT live streaming usually uploads via HTTP or RTMP using the Internet by unicast protocol.

– OTT live streaming delivery usually delivers content via HTTP. Because the prevalent HTTP version 1.1 does not support multicast, OTT live streaming service only supports unicast through the CDN.

– OTT live streaming service usually uses adaptive HTTP streaming to deliver media content, which relies on HTTP on top of TCP, and therefore only supports unicast streaming. Meanwhile, because TCP is reliable, the OTT live streaming system does not need to consist of an error recovery function unlike the IPTV live streaming system.

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